AMENDMENTS TO THE CLAIMS:

This listing of claims will replace all prior versions and listings of claims in the application:

1. (Previously Presented) A method for processing audio signals comprising:

quantizing the audio signals in spectral lines into quantized data in a plurality of sub-bands in an order of most significant bits to least significant bits;

determining, according to a psychoacoustic model, a plurality of scale factors corresponding to the plurality of sub-bands according to respective noise tolerance of each of the sub-bands;

for each scale factor that exceeds a threshold value, bit shifting the quantized data in the corresponding sub-band by the scale factor, wherein the threshold value is predetermined according to a desired noise tolerance level;

coding the quantized data; and

truncating the quantized data.

2. (Original) The method of claim 1 further comprising:

de-shifting the coded data;

de-quantizing the coded data; and

decoding the coded data.

3. (Original) The method of claim 2 further comprising:

amplifying the quantized data with the respective scale factors; and

de-amplifying the decoded data with the respective scale factors.

4. (Currently Amended) The method of claim 2 further comprising determining a difference ~~of~~ between the quantized data and the de-quantized data.

5. (Original) The method of claim 1 further comprising coding the quantized data in a base layer and an enhancement layer.

6. (Original) The method of claim 5 further comprising truncating the quantized data in the enhancement layer ~~up~~ to comply with respective layer size limits.

7. (Original) The method of claim 1 further comprising one of Huffman coding, run length (RL) coding or arithmetically coding the quantized data.

8. (Previously Presented) The method of claim 1, wherein the scale factor of a sub-band is determined based upon an original spectral energy level, a common scale factor, and band scale factor values of the sub-band.

9. (Original) The method of claim 1 further comprising converting the audio signals from a time domain to a frequency domain.

10. (Original) The method of claim 2 further comprising converting the decoded data from a frequency domain to a time domain.

11. (Currently Amended) A scale factor based bit shifting (SFBBS) system having an encoder and decoder to process audio signals, comprising:

an encoder including

a quantizer to quantize the audio signals in spectral lines into quantized data in a plurality of sub-bands in an order of most significant bits to least significant bits;

a psychoacoustic model to determine a plurality of scale factors corresponding to the plurality of sub-bands according to respective noise tolerance of each of the sub-bands;

a coder to code the quantized data;

a de-quantizer to de-quantize the quantized data;

a subtractor to take a difference ~~of~~ between the quantized data and the de-quantized data;

a bit shifter to shift the difference by the corresponding scale factor in each of the sub-bands in which the corresponding scale factor exceeds a threshold value, wherein the threshold value is predetermined according to a desired noise tolerance level; and

a bit slicer to code and truncate the difference.

12. (Previously Presented) The system of claim 11 further comprising:

a decoder having

a scale factor decoder to decode the scale factors;

a spectrum decoder to decode the quantized data;

a de-shifter to de-shift the coded data; and

a decoder to decode the coded data.

13. (Previously Presented) The system of claim 11, the encoder further comprising a filter to convert the quantized data from a time domain to a frequency domain.

14. (Previously Presented) The system of claim 12, the decoder further comprising a filter to convert the decoded data from a frequency domain to a time domain.

15. (Previously Presented) The system of claim 12, the decoder further comprising an adder to add the decoded data.

16. (Currently Amended) The system of claim 12 wherein the quantized data are amplified <u>with the respective scale factors, </u>and[[,]] the decoded data <u>are </u>de-amplified[[,]] with the respective scale factors.

17. (Previously Presented) The system of claim 11 further comprising one of a run length (RL) encoder, Huffman encoder or bit slice arithmetic encoder to code the quantized data.

18. (Original) The system of claim 11 being implemented in an additive fine granularity scalability (FGS) structure.

19. (Original) The system of claim 11 wherein the least significant bits are discarded after the bit shifting.

20. (Currently Amended) The system of claim 11 wherein the quantized difference is coded in a base layer and an enhancement layer, and the quantized difference in the enhancement layer is truncated ~~up~~ to comply with respective layer size limits.

21. (Previously Presented) A method for processing audio signals comprising:

quantizing the audio signals in spectral lines into quantized data in a plurality of sub-bands in an order of most significant bits to least significant bits;

determining, according to a psychoacoustic model, a plurality of scale factors corresponding to the plurality of sub-bands according to respective noise tolerance of each of the sub-bands;

for each scale factor that exceeds a threshold value, bit shifting the quantized data in the corresponding sub-band by the scale factor, wherein the threshold value is predetermined according to a desired noise tolerance level;

coding the quantized data in a base layer; and

truncating the quantized data.

22. (Original) The method of claim 21 further comprising:

de-shifting the coded data;

de-quantizing the coded data; and

decoding the coded data.

23. (Original) The method of claim 21 further comprising discarding the least significant bits after the bit shifting.

24. (Currently Amended) The method of claim 21 further comprising:

coding the quantized data in a base layer and an enhancement layer; and

truncating the quantized data in the enhancement layer ~~up~~ to comply with respective layer size limits.

25. (Original) The method of claim 21 further comprising one of Huffman coding, arithmetically coding or run length (RL) coding the quantized data.

26. (Previously Presented) The method of claim 21, wherein the scale factor of a sub-band is determined based upon an original spectral energy level, a common scale factor, and band scale factor values of the sub-band.

27. (Original) The method of claim 21, the method being implemented in an additive fine granularity scalability (FGS) structure.

28. (Previously Presented) A scale factor based bit shifting (SFBBS) system having an encoder and decoder to code and decode, respectively, audio signals, wherein the encoder comprises:

a quantizer to quantize the audio signals in spectral lines into quantized data in a plurality of sub-bands in an order of most significant bits to least significant bits;

a psychoacoustic model to determine a plurality of scale factors corresponding to each of the sub-bands according to respective noise tolerance of each of the sub-bands;

a bit shifter to shift the quantized data by the corresponding scale factor in each of the sub-bands in which the corresponding scale factor exceeds a threshold value, wherein the threshold value is predetermined according to a desired noise tolerance level; and

a bit slicer to code and truncate the quantized data.

29. (Previously Presented) The system of claim 28, wherein the decoder comprises:

a scale factor decoder to decode the scale factors;

a spectrum decoder to decode the quantized data;

a de-shifter to de-shift the coded data; and

a decoder to decode the coded data.

30. (Original) The system of claim 28 being implemented in MPEG-4 bit slice arithmetic coding (BSAC).

31. (Previously Presented) A method for processing audio signals, comprising:

quantizing the audio signals in spectral lines into quantized data in a plurality of sub-bands in an order of most significant bits to least significant bits;

determining, according to a psychoacoustic model, a plurality of scale factors corresponding to each of the sub-bands according to respective noise tolerance of each of the sub-bands;

de-quantizing the quantized data;

for each scale factor that exceeds a threshold value, bit shifting the quantized data in the corresponding sub band by the scale factor, wherein the threshold value is predetermined according to a desired noise tolerance level; and

coding and truncating the quantized difference.

32. (Original) The method of claim 31 further comprising:

de-shifting the coded data; and

decoding the coded data.

33. (Original) The method of claim 32 further comprising:

amplifying the quantized data with the respective scale factors; and

de-amplifying the decoded data with the respective scale factors.

34. (Original) The method of claim 31 further comprising one of Huffman coding, run length (RL) coding or arithmetically coding the quantized data.

35. (Original) The method of claim 31 wherein the least significant bits, after the bit shifting, are discarded.

36. (Previously Presented) A scale factor based bit shifting (SFBBS) processor for processing audio signals in an order of most significant bits to least significant bits, the processor comprising:

a psychoacoustic module to determine a plurality of scale factors corresponding to a plurality of spectral sub-bands according to respective noise tolerance of each of the sub-bands;

a bit shifter to shift the processed audio signals by the corresponding scale factor in each of the spectral sub-bands in which the corresponding scale factor exceeds a

threshold value, wherein the threshold value is predetermined according to a desired noise tolerance level; and

a bit slicer to code and truncate the processed audio signals.

37. (Previously Presented) The processor of claim 36 further comprising a quantizer to quantize the processed audio signals.

38. (Previously Presented) The processor of claim 36 further comprising:

a quantizer to quantize the processed audio signals;

a de-quantizer to de-quantize the processed audio signals; and

a subtractor to take a difference between the quantized audio signals and the de-quantized audio signals.

39-40. (Canceled).